REMARKS

Reconsideration of the application is requested.

Claims 1-10 remain in the application. Claims 1-10 are subject to examination. Claim 1 has been amended. An RCE has been concurrently filed.

Under the heading "Claim Rejections – 35 USC § 103" on page 3 of the above-identified Office Action, claims 1-10 have been rejected as being obvious over U.S. Publication No. 2003/0133565 to Chang et al. in view of U.S. Publication No. 2002/0016161 to Dellien et al. under 35 U.S.C. § 103.

In the third paragraph of page 2 of the advisory action, the Examiner has stated that: "The end result from modification for echo reduction produces or generates an uplink data in compressed state. In other words, the uplink data in compressed state is the end result or output that is performed or generated from using the results of the analysis of the downlink data to modify for echo reduction."

Claim 1 has been amended to specify that the uplink data is in <u>a compressed state prior to being modified for echo reduction</u>. Support for the change can be found by referring to claim 1 as originally presented. The Examiner may also refer to the first and last paragraphs on page 2 of the translated specification.

The Examiner can no longer interpret the claim language in the manner discussed in the third paragraph of page 2 of the advisory action.

The Examiner has compared the far-end speech from the speaker 121 of Chang et al. with the claimed uplink data and the signal S(n) of Chang et al. with the claimed downlink data. The Examiner has alleged that Dellien et al. would have suggested compressing the signal S(n) of Chang et al. (the uplink data).

Applicants respond to the Examiners allegation by pointing out that compressing the signal S(n) of Chang et al. would have resulted in an inoperable device and that therefore one of ordinary skill in the art would not have obtained a suggested to make the change. The signal S(n) has to be a digitized waveform, namely a digitized speech waveform, so that it can be summed with the echo error signal e1(n) from the adaptive filter 124 in order to obtain the echo free input signal d(n) that is then fed to the voice encoder 125 in order to be compressed. <u>This type of process cannot be performed with compressed speech signals</u>.

In a compressed form, the speech is represented as parameters and coefficients like codebooks and excitation pulse delays, and resonants of the vocal tract. This is information that characterizes the speech. Due to the high level of abstraction, speech is compressed and the data rate is reduced. An example is MP3 compression. But compressed speech is not a signal in the

sense of a digitized waveform like the one used in Chang et al. The parameters can be used to reconstruct the speech at the receiver, but they do not represent any digitized samples of a signal as used by Chang et al. and can therefore not be filtered or treated like a signal. In order to perform echo cancellation in Fig. 1, Chang et al. uses a digitized waveform in 124, 126 which is generated by an ADC 127. This speech signal is similar to a telephone line. Voice compression/coding only happens after this process in voice encoder 125. If we use the MP3 analogy, you record a voice or copy a CD and the result is a .wav file containing digitized speech and where you can do all kinds of processing, filtering etc. Only after this you use an encoding/compression program to convert the .wav to MP3, WMA or iPod format. The compressed speech cannot be filtered in the way Chang et al. teaches.

Applicants also point out that Dellien et al. teach a compression method that is used to reduce the memory requirements for a voice memo function in a mobile telephone. There is no teaching or suggestion that the compression method could be used for compressing speech during a real-time telephone conversation. Furthermore, if one of ordinary skill in the art were to use the compression method taught by Dellien et al. in the device of Chang et al., <u>it would have been applied in the voice encoder 125 of Chang et al. after the echo cancelation process</u>. The signal S(n) would not have been compressed.

Applicants believe they have made their point, but will now nevertheless provide more detailed comments.

<u>Chang et al. use digitized waveforms to perform echo cancellation:</u>

- In [0005] Chang et al. teaches that the echo cancellation is performed in 8 KHz pulse code modulation, which is digitized speech with a sampling rate of 8 KHz (like a telephone line) and which is completely different from compressed speech containing e.g. codebooks.
- In [0006] Chang et al. teach that speech is digitized in the ADC 127 with 8 kHz to form a digital input signal S(n), which is a digitized waveform, but is not compressed speech.
- In [0007] Chang et al. teaches that the signals are discrete-time versions after ADC, which is a digitized waveform. Voice encoding/compression is mentioned as a separate processing step 125 that is performed after the echo cancellation.

<u>Chang et al. teach methods that cannot be performed on encoded/compressed speech:</u>

- In [13] Chang et al. teaches detecting double talk by monitoring voice energy in a first frequency band. This is not possible for compressed speech since the compressed speech is represented by special parameters and coefficients like codebooks. There is no frequency band that can be extracted to be used for voice energy monitoring. This can only be done for digitized and uncompressed speech.
- Voice energy monitoring is done e.g. by adding up the quadrature of digitized samples. This is not possible for compressed speech since there are not

digitized speech samples, but only coefficients and parameters which do not allow one to extract any energy information.

- Chang et al. teaches using an adaptive filter for producing an echo signal. However, an adaptive filter can only be applied to digitized voice samples. An adaptive filter cannot be applied to the parameters of compressed speech that contain e.g. codebooks. This is completely impossible.
- Also in [21] Chang et al. talks about signals in different frequency bands - meaning digitized waveforms that are used for filtering and analysis. This is not applicable and is completely unrelated to parameters and coefficients of compressed speech.

Applicants therefore believe that the invention as defined by claim 1 would not have been suggested. Applicants believe that the reasons given above with regard to claim 1 are also applicable to claim 10.

It is accordingly believed to be clear that none of the references, whether taken alone or in any combination, either show or suggest the features of claims 1 or 10. Claims 1 and 10 are, therefore, believed to be patentable over the art. The dependent claims are believed to be patentable as well because they all are ultimately dependent on claim 1.

In view of the foregoing, reconsideration and allowance of claims 1-10 are solicited.

In the event the Examiner should still find any of the claims to be unpatentable, counsel would appreciate receiving a telephone call so that, if possible, patentable language can be worked out.

Petition for extension is herewith made. The extension fee for response within a period of one month pursuant to Section 1.136(a) in the amount of $130.00 in accordance with Section 1.17 is enclosed herewith.

Please charge any other fees that might be due with respect to Sections 1.16 and 1.17 to the Deposit Account of Lerner Greenberg Stemer LLP, No. 12-1099.

Respectfully submitted,

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